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[56] 参考文献
US5375121A 1994.12.20
WO00/02410A2 2000.1.13
"NARROWBAND SERVICES OVER ATM-NETWORKS: EVALUATION OF TRUNKING METHODS". SINGER J ET AL. ISS97 WORLD TELECOMMUNICATIONS CONGRESS, Vol. 2. 1997
"VOICE OVER ATM: A HYBRID TDM/ATMDESIGN APPROACH". LAURENCE M ET AL. ELECTRONIC ENGINEERING, Vol. 71 No. 869. 1999
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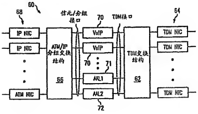
话音分组交换机和交换系统

[57] 摘要

一种话音分组交换系统包括可操作来在传送话音数据分组的多个信道之间交换的至少一个分组交换结构，耦合到该分组交换结构并且可操作来在所述话音数据分组与多个话音数据流之间转换的至少一个话音业务模块，以及耦合到所述至少一个话音业务模块的时分复用交换结构。



1. 一种话音分组交换机，包括：

用于在多条 ATM 传输链路上接收和发送数据分组和用于从 ATM 数据分组中提取数据并且将提取的数据插入到预定协议的数据帧中的多个 ATM 分组处理电路；以及

时隙交换机，其被耦合到所述多个 ATM 分组处理电路并且被配置来进行数据帧和数据的 DS0 级别和其它每话音信道级别的交换，所述数据帧是从 ATM 分组处理电路接收的数据帧，所述数据是通过多条 PCM 传输链路接收和发送的复用 PCM 数据；

其中所述话音分组交换机被配置作为一个 PSTN 网络交换单元以及桥接 PSTN 网络和基于分组的网络的网关来操作。

2. 根据权利要求 1 所述的话音分组交换机，其中所述多个 ATM 分组处理电路包括用于接收和发送 AAL1 信元的 AAL1 分组处理电路。

3. 根据权利要求 1 所述的话音分组交换机，其中所述多个 ATM 分组处理电路包括用于接收和发送 AAL2 信元并且用于从 AAL2 信元中提取数据并且将提取的数据插入到 HDLC 数据帧中来由所述时隙交换机交换的 AAL2 分组处理电路。

4. 根据权利要求 1 所述的话音分组交换机，其中所述多个 ATM 分组处理电路包括用于接收和发送 VoIP 数据分组并且用于从所述 VoIP 数据分组中提取数据并且将所提取的数据插入到 HDLC 数据帧中来由所述时隙交换机交换的 VoIP 分组处理电路。

5. 根据权利要求 1 所述的话音分组交换机，还包括耦合在所述多个 ATM 分组处理电路和用于在 ATM 数据分组与复用 PCM 数据之间转换的时隙交换机之间的至少一个数字信号处理电路。

6. 根据权利要求 5 所述的话音分组交换机，其中所述至少一个数字信号处理电路包括至少一个回波消除电路，用于消除来自被所述时隙交换机所交换的话音信道的回波。

7. 根据权利要求 5 所述的话音分组交换机，其中所述至少一个数字信号处理电路包括至少一个话音压缩电路，用于压缩被用来在所述时隙交换机所交换的话音信道中编码话音的数据。

8. 一种交换系统，包括：

至少一个被配置来处理通过至少一条分组传输链路接收和发送的

非 TDM 数据分组的分组处理电路；以及

一个交换装置，该交换装置具有：

至少一个第一信道，被配置来通过所述至少一条分组传输链路接收和发送非 TDM 数据分组；以及

至少一个第二信道，被配置来通过所述至少一条非分组传输链路接收和发送复用 TDM 数据；

其中所述交换装置被配置来在所述至少一个第一信道和至少一个第二信道之间进行 DS0 级别和其它每话音信道级别的交换；

其中所述交换系统被配置来作为 PSTN 网络交换单元和桥接 PSTN 网络和基于分组的网络之间的网关进行操作。

9. 根据权利要求 8 所述的交换系统，其中所述至少一个分组处理电路用于提取通过所述至少一条分组传输链路接收的非 TDM 数据分组并且使用一个数据链路协议发送到所述交换装置。

10. 根据权利要求 8 所述的交换系统，其中所述至少一个分组处理电路用于提取通过至少一条分组传输链路接收的压缩话音数据并且使用一个数据链路协议发送到交换装置进行交换。

11. 根据权利要求 8 所述的交换系统，其中所述至少一个分组处理电路用于提取通过至少一条分组传输链路接收的 AAL2 话音数据并且使用高级数据链路控制协议发送到交换装置进行交换。

12. 根据权利要求 8 所述的交换系统，其中所述至少一个分组处理电路用于提取通过至少一条分组传输链路接收的 VoIP 话音数据并且使用高级数据链路控制协议发送到交换装置进行交换。

13. 根据权利要求 8 所述的交换系统，其中所述至少一个分组处理电路用于提取从所述交换装置接收的数据并且将所提取的数据插入到分组中以便通过至少一条分组传输链路进行发送。

14. 根据权利要求 8 所述的交换系统，其中所述至少一个分组处理电路用于提取从所述交换装置接收的话音数据并且将所提取的话音数据插入到 AAL 信元中以便通过至少一条分组传输链路进行发送。

15. 根据权利要求 8 所述的交换系统，其中至少一个分组处理电路用于提取从所述交换装置接收的话音数据并且将所提取的话音数据插入到 VoIP 分组中以便通过至少一条分组传输链路进行传输。

16. 根据权利要求 8 所述的交换系统，其中所述交换装置是时隙

交换机交换装置。

17. 根据权利要求 8 所述的交换系统，其中所述交换装置用于交换复用的未压缩话音数据。

18. 根据权利要求 8 所述的交换系统，其中所述交换装置用于交换 PCM 数据。

19. 根据权利要求 8 所述的交换系统，其中所述至少一条分组传输链路包括多条分组传输链路，所述多条分组传输链路包括复用的 VoIP 数据传输链路和复用的 AAL2 数据传输链路，所述复用的 VoIP 数据传输链路用于向和从所述交换装置传输包封有话音数据的 IP 分组，所述复用的 AAL2 数据传输链路用于向和从所述交换装置传输 AAL2 分组，该 AAL2 分组承载了包封有话音数据的复用 ATM 信元。

20. 根据权利要求 8 所述的交换系统，其中所述至少一个分组处理电路包括至少一个用于与所述交换装置的至少一条 ATM 传输链路接口的 AAL1 分组处理电路，用于处理承载了包封有话音数据的 ATM 信元的 AAL1 分组。

21. 根据权利要求 8 所述的交换系统，其中所述至少一个分组处理电路包括至少一个用于与所述交换装置的至少一条 ATM 传输链路接口的 AAL2 分组处理电路，用于处理承载了包封有话音数据的复用 ATM 信元的 AAL2 分组。

22. 根据权利要求 8 所述的交换系统，其中所述至少一个分组处理电路包括至少一个用于与至少一条 IP 传输链路接口的 VoIP 分组处理电路，用于处理包封有话音数据的 IP 分组。

23. 根据权利要求 8 所述的交换系统，还包括至少一个数字信号处理电路，该数字信号处理电路耦合在所述至少一个分组处理电路与所述交换装置之间并且用于在对应于所述至少一条分组传输链路的第一数据格式与对应于所述至少一条非分组传输链路的第二数据格式之间进行转换。

24. 根据权利要求 8 所述的交换系统，还包括至少一个数字信号处理电路，该数字信号处理电路耦合在所述至少一个分组处理电路与所述交换装置之间并且用于在对应于所述至少一条分组传输链路的压缩数据格式与对应于所述至少一条非分组传输链路的非压缩数据格式之间进行交换。

25. 根据权利要求24所述的交换系统，其中所述至少一个数字信号处理电路包括至少一个回波消除电路，用于消除来自被所述交换装置所交换的话音信道的回波。

26. 根据权利要求24所述的交换系统，其中所述至少一个数字信号处理电路包括至少一个话音压缩电路，用于压缩被用来对所述交换装置所交换的话音数据进行编码的数据。

话音分组交换机和交换系统

技术领域

本发明涉及电信设备，并且更具体而言，本发明涉及话音分组交换系统和方法。

背景技术

电信网发展的当前趋势集中在利用新兴的基于分组的数据网来代替传统的公共交换电话网（PSTN）。存在多个对于这一趋势的贡献因素。数据业务量大大超过话音业务量并且这个差距继续增加而没有衰减。研究表明话音将只构成总电信业务量的非常小的一部分。此外，数据网固有地对于支持各种话音和数据业务更灵活并且具有支持目前通过PSTN传送的所有业务的技术潜能。

需要几年时间来解决技术问题，从而数据网能够具有PSTN的较高质量和可靠性。同时，PSTN与快速发展的数据网将继续共存。为了便于话音技术从PSTN向分组网移动并且使两种网络对于话音业务互操作，网络运营商正在采用话音媒体网关。话音媒体网关将话音信号在时分复用（TDM）脉冲编码调制（PCM）和数据分组格式之间转换，所述数据分组格式或者是异步转移模式（ATM）或者是互联网协议上的话音（VoIP）。因此，话音媒体网关目前用于将数据网与PSTN桥接，但是不执行必要的交换功能。尽管是必要的，但是这种类型的转接网是不经济的，这是因为它引入了加倍的网元并且增加了操作和维护网络的成本。

发明内容

为解决以上技术问题，本发明的话音分组交换机包括：用于在多条ATM传输链路上接收和发送数据分组和用于从ATM数据分组中提取数据并且将提取的数据插入到预定协议的数据帧中的多个ATM分组处理电路；以及时隙交换机，其被耦合到所述多个ATM分组处理电路并且被配置来进行数据帧和数据的DS0级别和其它每话音信道级别的交换，所述数据帧是从ATM分组处理电路接收的数据帧，所述数据是通过多条PCM传输链路接收和发送的复用PCM数据；其中所述话音分组交换机被配置作为一个PSTN网络交换单元以及桥接PSTN网络和基于分组

的网络的网关来操作。

本发明还提供一种交换系统，包括：至少一个被配置来处理通过至少一条分组传输链路接收和发送的非TDM数据分组的分组处理电路；以及一个交换装置，该交换装置具有：至少一个第一信道，被配置来通过所述至少一条分组传输链路接收和发送非 TDM 数据分组；以及至少一个第二信道，被配置来通过所述至少一条非分组传输链路接收和发送复用 TDM 数据；其中所述交换装置被配置来在所述至少一个第一信道和至少一个第二信道之间进行 DS0 级别和其它每话音信道级别的交换；其中所述交换系统被配置来作为PSTN网络交换单元和桥接PSTN网络和基于分组的网络之间的网关进行操作。

根据本发明，话音分组交换机提供端到端话音交换，而与主叫和被叫方使用的编码、复用或者传输协议以及沿着话音连接路径的交换机间中继线无关。话音分组交换还利用所有的话音编码和压缩方案、物理层传输协议、接入协议和中继协议来进行操作。本发明的一个重要方面是分离在分组交换网上交换话音中的三个关键功能：分组处理、数字信号处理和交换。本发明的话音分组交换允许以对于高密度应用最成本有效的方式来实现每个关键功能，诸如在中央局中。此外，可以利用互相独立的最先进水平的技术来实现每个功能。从话音分组交换机中除去回波消除和话音压缩/解压缩功能所带来的另一个优点是提高的话音保真度和回波消除质量。此外，还消除了在话音传输中与抖动缓存相关的严重的端到端延迟。关键在于消除沿着话音连接路径的中间交换机上的话音压缩/解压缩循环。

在本发明的一个实施例中，交换系统包括与至少一条分组传输链路接口的至少一个分组处理电路。该交换系统还包括一个耦合到所述至少一个分组处理电路的交换结构，从而该分组交换结构可操作来在通过至少一条分组传输链路接收和发送数据的信道与通过由所述交换结构接口的至少一条非分组传输链路接收和发送数据的信道之间交换。

在本发明的另一个实施例中，交换系统包括至少一个与第一数据业务量接口的分组处理电路，所述第一数据业务量是被使用第一传输协议接收和发送的以第一协议的形式。所述交换系统还包括一个耦合到至少一个分组处理电路的交换结构，从而该交换结构可操作来在耦

合到第一数据业务量的信道与耦合到第二数据业务量的信道之间交换，所述第二数据业务量是被使用由所述交换结构接口的第二传输协议接收和发送的以第二协议的形式。

在本发明的另一个实施例中，话音分组交换机包括多个可操作来通过多条 ATM 传输链路接收和发送数据分组并且可操作来从 ATM 数据分组中提取数据并且将所提取的数据插入到预定协议的数据帧中的多个 ATM 分组处理电路。所述话音分组交换机还包括一个耦合到所述多个 ATM 分组处理电路并且可操作来交换从中接收的数据帧以及通过多条 PCM 传输链路接收和发送的复用的 PCM 数据的时隙交换机。

在本发明的另一个实施例中，话音分组交换机包括可操作来通过多条 IP 传输链路接收和发送数据分组并且可操作来从 IP 数据分组中提取数据并且将所提取的数据插入到预定协议的数据帧中的多个 VoIP 分组处理电路。所述话音分组交换机还包括一个耦合到所述多个 VoIP 分组处理电路并且可操作来交换从中接收的数据帧以及通过多条 PCM 传输链路接收和发送的复用的 PCM 数据的一个时隙交换机。

在本发明的另一个实施例中，一种交换话音分组的方法包括步骤：接收通过至少一条分组传输链路发送的数据分组，处理所述数据分组，并且将所述数据分组转换成为预定数据链路协议的数据帧。该数据帧被在一个交换结构的一个输入信道中接收，并且被输出到该交换结构的输出信道。然后，数据帧被转换成为数据分组，并且被通过至少一条分组传输链路来发送。

附图说明

为了更完整地理解本发明及其目的和优点，现在参考附图来描述本发明，在附图中：

图 1 是采用本发明的话音分组交换系统的演化电信网的简化框图；

图 2 是根据本发明教导的话音分组交换系统的实施例的简化结构框图；

图 3 是根据本发明教导的话音分组交换系统的替代实施例的简化框图；

图 4 是根据本发明教导的话音分组交换系统的另一个实施例的简化框图；并且

图 5 是所述话音交换系统的一对一（any-to-any）功能以及利用

相同的话音压缩编码方案对业务量进行分组的方法的图示。

具体实施方式

通过参考附图1到5来最好地理解本发明的优选实施例及其优点，类似的编号用于各图中类似和相应的部分。

图1是采用本发明的话音分组交换系统12－14的演化电信网10的简化框图。演化网络10是操作在用于话音业务的公共交换电话网（PSTN）16以及用于数据业务的异步转移模式网络（ATM）18和互联网协议网络（IP）20中的当前和下一代设备的聚结。ATM是高带宽、低延迟、面向连接的类似分组的交换和复用技术，用于支持数据、话音数据、视频数据、图像数据、多媒体数据业务量的传输。ATM网络18采用诸如ATM交换机22、数字用户线接入复用器（DSLAM）24和综合接入设备（IAD）26的设备。ATM网络18还包括本发明的一个或多个话音分组交换机12用于在ATM网络中交换单独的复用的话音流。这里将话音流定义为在相同的话音连接中从一方到另一方的单向连续话音信号。

ATM网络18经由话音媒体网关(MG)30和31耦合到PSTN 16。PSTN采用诸如类型5交换机32、类型4交换机（未示出）、数字回路载波34和信道处理单元（CB）36的设备来路由选择和递送话音和数据业务量。PSTN 16还经由话音媒体网关40耦合到IP网络20以便在TDM PCM与IP之间转换话音信号。除了本发明的话音分组交换系统14之外，IP网络20还采用诸如路由器42、DSLAM 44和IAD 46的设备。使用本发明的话音分组交换系统，话音业务量能够被通过诸如目前被支持的IP和ATM协议的任何数据分组协议（VoX）传送。

可以看到话音分组交换机可以作为两种完全不同形式的网络之间的接口设备，诸如ATM和IP网18和20之间的话音分组交换机14。话音媒体网关在演化网络10中的存在是由于它们集成到电信网中来解决引入话音分组交换系统之前的话音-数据接口问题。在配置话音分组交换系统的情况下，当网络配置逻辑以及资金设备调度允许时，话音媒体网关最终可以被逐步淘汰。

本发明的话音分组交换系统可操作来执行一对一交换，以便能够在任何编码和传输方案之间交换话音流。话音分组交换机可操作来终接入站信道，提取其中传送的话音信号并且将它放回到匹配出站信道

中，而与入站和出站信道使用相同还是不同的编码和传输方案无关。此外，甚至在很大程度上经济地执行这些功能。因为在各话音流被复用在一条单独连接上的方式中，TDM（时分复用）到 AAL1（ATM 适配层类型 1）与 AAL2（ATM 适配层类型 2）到 VoIP（IP 上的话音）之间存在基本的差别，所以本发明的话音分组交换机区分它们并且对于它们执行不同的操作。每个 TDM 信道或者 AAL1 连接只传送一个话音流，而每个 AAL2 连接可以传送 240 个同时的话音流，并且每条 VoIP 连接甚至可以传送更多。本发明的话音分组交换系统可操作来对于话音流进行解复用并且根据每个话音对话的路径来将它们单独地交换到其相应的出站信道，然后将共享相同的下一个中继交换机的所有出站话音流再复用到相同的出站 AAL2 或者 VoIP 连接。这样，可以将话音应用从对于最终用户关系很小或者很不重要的编码和传输方案中分离。最后，作为用于网络通信公司的中央局交换机，本发明的话音分组交换机可操作来随着话音端口密度的增加而经济地按比例增加。

技术的当前发展水平不包括满足以上列举的需求的任何设备。例如，媒体网关只执行或者 TDM 与 ATM 之间的话音转换或者 TDM 与 VoIP 之间的话音转换，而很少执行以上两种转换。此外，媒体网关不执行交换功能。目前，没有 IP 路由器或者 ATM 交换机在话音流被复用时执行单独的话音流交换。目前的所有 ATM 交换机都只执行 ATM 格式的交换，并且 IP 路由器只在不同的 IP 端口之间交换 IP 分组。不执行话音流的复用。此外，IAD 与 DLC 只在 IP 或者 ATM 与 TDM 之间转换话音流，并且典型地不执行任何交换功能。下面参考附图 2－5 可以看到，本发明的话音分组交换系统和方法提供了将话音和数据通信聚合到未来的以数据为中心的电信网中的唯一解决方案。

图 2 是根据本发明教导的话音分组交换系统 60 的实施例的简化基本结构框图。话音分组交换系统 60 包括 TDM 交换结构 62，它经由 TDM 网络接口控制器（NIC）64 耦合到 TDM 传输链路。TDM 交换结构 62 是具有 DS0 或者 64Kbps 带宽的信道尺寸的时隙交换机。ATM/IP 交换结构 66 可操作来交换可变长度分组。耦合到 ATM/IP 交换结构 66 的网络接口控制器 68 可操作来终接 IP 或者 ATM 链路。诸如 VoIP 70、AAL1 和 AAL2 的话音服务模块是可操作来执行接口、分组处理和可选地话音压缩和回波消除的电路或者芯片。AAL1 支持类型 A 业务量，类型 A 业务

量是面向连接的固定比特率（CBR）的依赖于时间的业务量，诸如未压缩的数字化话音和视频数据。AAL2 支持类型 B 业务量，类型 B 业务量是面向连接的可变比特率（VBR）的需要信源和信宿之间的精确定时的同步业务量，诸如被压缩的话音和视频业务量。尽管这些话音业务模块被在图 2 中示为单独的逻辑块，但是它们也可以被集成和实现在一个单独的电路板或者各种配置中。交换机 60 可操作来执行话音和数据格式与速度之间的一对一交换。

图 3 是根据本发明教导的话音分组交换系统 80 的实施例的简化框图。话音分组交换系统 80 是图 2 所示基本结构的简单易懂的实现。话音分组交换系统 80 包括分别接口和处理 AAL1、AAL2 和 VoIP 分组的 AAL1 分组处理（PP）电路或者芯片 82、AAL2 分组处理电路或者芯片 84 和 VoIP 分组处理电路或者芯片 86。AAL1 分组处理电路 82 被经由回波消除电路或者芯片 90 和脉冲编码调制链路 92 耦合到时隙交换机 88 或者 TDM 交换结构。AAL2 分组处理电路 84 经由回波消除电路 94、话音压缩电路或者芯片 96 和脉冲编码调制/高级数据链路控制（PCM/HDLC）链路 98 耦合到时隙交换机 88。HDLC 是用于点到点和点到多点通信的链路层协议标准。HDLC 将分组数据封装到一个帧中，该帧具有包括诸如错误控制机制的各种控制信息的帧头部和尾部。HDLC 变体包括帧中继、平衡型链路接入规程（LAP－B）、链路接入过程数据信道（LAP－DC）、点到点协议（PPP）和同步数据链路控制（SDLC）。

使用数据链路层协议，每条话音连接一个比特流 HDLC 信道被用于通过时隙交换机 88 来互连两个 AAL2 和 VoIP 分组处理电路。封装在 AAL2 信元或者 VoIP 分组中的已编码话音帧（或者小分组）被提取并且被放到 HDLC 帧中以便通过时隙交换机 88 交换。交换之后，这些话音帧被放回到出站 AAL2 信元或者 VoIP 分组中。用于寂静检测的寂静插入描述符（SID）消息可以使用不同的用户到用户指示（UUI）透明地传送通过标准 ATM 接口和时隙交换机 88 之间的 AAL2 分组处理电路 84，以便传播通过网络。AAL2 分组处理电路 84 不执行业务特定会聚子层（SSCS）功能，所以 SID 消息和话音分组被同样对待。HDLC 信道开销包括用于 AAL2 头的两个字节、用于 HDLC 头的两个字节以及比特填充，同时总的 HDLC 信道原始带宽被限制为 64kbps。如果 PCM 被用于通过 AAL1 传送话音流，则 PCM 流能够在它们自己的 DS0TDM 格式中传送通过

来代替使用 HDLC 信道。在这个实施例中选择 HDLC 协议，这是因为它被广泛实现并且许多商业现货供应的硬件和软件都是容易获得的。如果额外的利益能够被给予，则其它恰当的协议也可以被使用。在压缩之后，话音流不高于 40Kbps。话音有效负荷、HDLC 开销和 AAL2/VoIP 开销的总和在 64Kbps 之下。这个至关紧要的观察资料迫使使用时隙交换机 88 作为统一的交换结构来满足所有的话音流交换需求。

在操作中，最好是在 PCM 中进行一对 TDM 业务量之间的交换。一对 ATM AAL1 之间的交换最好是在 PCM 中进行。具有相同的自适应差分脉冲编码调制（ADPCM）编码子类型的任何 AAL2/VoIP 到 AAL2/VoIP 交换的组合都可以被使用直接 PCM 或 HDLC 来交换。如果直接 PCM 被选择，则只有在已编码话音抽样的每个字节中的固定子集的比特被填充、处理和用于播放。这个选项需要抖动缓存的激活。如果 HDLC 被选择，则不需要抖动缓存。抖动缓存被用于在分组逐段通过电信网传播时消除随机分组延迟变化。因为来自同一话音对话的两个连续信元或分组的到达之间的时间间隔的随机变化，所以需要每个交换机设想最坏情况的延迟并且提供大的抖动缓存器。结果，累积的传输延迟对于跨越许多中继段的话音连接是非常重大的，对于收费的长途呼叫典型地就是这样。此外，这个大延迟使得难以进行满意的回波消除，并且严重削弱话音对话的主观质量或者逼真度。因此，HDLC 路径优于 PCM。具有相同的非 ADPCM 压缩编码的任何 AAL2/VoIP 到 AAL2/VoIP 的组合都最好被使用 HDLC 来交换。最好在交换之前首先将业务量转换到 PCM 来执行具有不同的压缩编码的 AAL2/VoIP 到 AAL2/VoIP 的任何组合。对于具有话音压缩的 AAL1/TDM 到 AAL2/VoIP，还最好在交换之前首先转换到 PCM。如果话音压缩根本未被使用，则交换最好在 PCM 中进行。目标是通过使用任何一个标准化的数据链路协议来封装和通过时隙交换机 88 在统一 PCM 信道上传送被压缩的话音流来如果可能则避免在中间中继段中到 PCM 的重复转换。被选择的数据链路协议的优选的属性包括可变长度话音小分组的接受、不超过 64Kbps 的总带宽以及错误检测和恢复能力，诸如 HDLC 和类似的协议。

如果不能获得使用标准化数据链路协议来在统一 PCM 信道上传送被压缩的话音流这一选项，则必须在每个中继段上执行到 PCM 的转换，从而在每个中继段都需要抖动缓存。此外，每个压缩和解压缩循环引

入了某个量的失真。随着话音连接通过多个交换机的这种失真的积累对于话音质量的额外损害是相当大的。另一个缺点是非常高的成本或者在与集成分组处理、回波消除和话音压缩芯片或模块相关的热消散上的限制。一个集成芯片也几乎不适合于中央局应用，这是因为高单元成本、低密度以及热消散。

考虑到朝向用于下一代网络的话音业务量的 AAL2/VoIP 的优势的当前趋势，本发明甚至更恰当。在所述网络中，在每个网络中继段执行重复的回波消除和话音压缩/解压缩的过程是非常无效率、不必要和不利的（归因于质量降低）。

图 4 是根据本发明教导的被从数字信号处理中分离的话音分组交换系统 120 的另一个实施例的简化框图。话音分组交换系统 120 不包括昂贵的数字信号处理电路或者芯片，并且因此不执行回波消除或者话音压缩。一条话音连接路径包括一个始发交换机、一个终接交换机和中间交换机。当话音压缩被使用时，它只是必须在始发交换机和终接交换机中进行而不必要在中间交换机中进行。对于从话音分组交换机中移出数字信号处理的另外促动因素是压缩话音的降低的经济和技术动机，这是因为网络带宽的惊人和继续的增加、数据业务量的爆炸性增长以及话音在整个业务混合中逐渐缩小的百分比。许多强制的数字信号处理功能以及可选择的话音压缩功能可以移动到用户端设备（CPE）中，这是因为包含分组处理、回波消除和话音压缩功能的集成芯片设备是容易获得的。这些芯片设备虽然不适合于中央局交换机，但是对于其中高集成功能是有价值的但是密度不是一个重要考虑事项的 CPE 应用则是理想的。

如图 4 所示，话音分组交换系统 120 包括 AAL1 分组处理电路 122、AAL2 分组处理电路 124 和 VoIP 分组处理电路 126。AAL1 分组处理电路 122 经由 PCM 数据链路 130 耦合到时隙交换机 128。AAL2 分组处理电路 124 经由 PCM/HDLC 数据链路 132 耦合到时隙交换机 128。VoIP 分组处理芯片 126 经由 PCM/HDLC 数据链路来 134 耦合到时隙交换机 128。业务量被以用于未压缩话音流的自然的 PCM 格式或者用于被压缩话音流的诸如 HDLC 的标准数据链路协议而呈现给时隙交换机 128 并且被该时隙交换机 128 交换。TDM 时隙交换机 128 是话音分组交换系统 120 的整体结构的组成部分以便于实现必要的类型 5 功能，诸如会议通话桥接、

用于法律实施行动的通信援助（CALEA）、操作员强插、音调产生、数字通告以及带内话音信令终接和拨号数字。

通过提供和/或配置网络，网络运营商可以在接入链路和中继线上建立独立的 AAL2 或者 VoIP 连接的组，每个组被为一个单独概况的话音信道而规定。一个概况是管理话音如何能够通过 IP 分组和 ATM AAL2 信元传送的方案。所述概况定义使用什么压缩编码、话音抽样或者帧的格式以及这些抽样或者帧如何被封装到 IP/RTP（互联网协议/实时传送协议）分组或者 ATM AAL2 信元中。一个话音连接建立请求将传送一个规定概况的选择的参数。在建立话音连接路径中，沿着所述路径的每个话音分组交换机上的控制软件将从具有与来自入站信道的那些相同概况的组中选择一个恰当的匹配出站信道。结果，话音连接的整个路径将使用相同的概况或者相同的压缩编码。图 5 示出了这个规定概念。

注意所建议的利用概况的划分不需要网络拓扑改变，它也不会必然影响话音呼叫的路由选择。分离或者划分只在逻辑连接等级上进行。在连接两个相邻的话音分组交换机的物理链路上，例如可以在接受话音呼叫之前提供两个或多个用于不同编码的 AAL2/VoIP 连接。

因为较少 DSP 的话音分组交换结构带来减少的脚印尺寸和/或增加的端口密度，所以与用于交换机的外壳和热消散相关的成本被大大降低。对于网络运营商的纯利益是在不动产、功耗以及空调费用方面的低得多的资本支出和正在进行的运营成本。大大简化的硬件结构还惊人地提高了整个交换系统的固有可靠性。因为与 DSP 相关的成本大大高于与分组处理和交换相关的成本，所以本发明的话音分组交换机获得以任何给定密度的最大的经济节约。

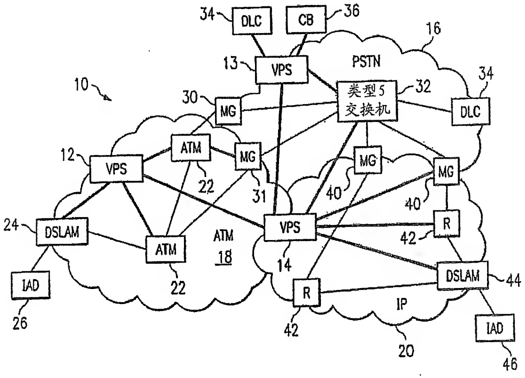
本发明的话音分组交换机获得端到端话音交换，而与主叫方和被叫方使用的编码、复用或者传输方案以及沿着话音连接路径的交换机间中继线无关。话音分组交换机还可利用所有话音编码和压缩方案、物理层传输协议、接入协议以及中继协议来操作。

本发明的一个重要方面是在分组交换网上交换话音中的三个主要功能的分离：分组处理、数字信号处理（回波消除和话音压缩）以及交换。在这样做时，本发明的话音分组交换机允许每个关键功能被以最成本有效的方式而为高密度应用来实现，诸如在中央局中。此外，

每个功能可以被利用最先进水平的技术而相互独立地实现。从话音分组交换机中消除 DSP 功能的另一个优点是提高的话音保真度和回波消除质量。此外，还消除了与抖动缓存相关的话音传输中的显著的端到端延迟。

本发明提供的成本利益可以被量化或者估计。对于一个具有 N 个端口的交换机，有 N-端口的接口电路和 N^2-端口的交换交叉点。业务提供商最关心的每个端口的成本是总成本除以 N。因此，每个端口成本是每端口接口成本和 N 个交叉点成本的总和。在本发明的优选实施例中，TDM 交换结构可以被以非常低的成本实现，这是因为它是简单和成熟技术的缩放。本发明首先减少了对于话音压缩/解压缩和回波消除的需要，然后提供了另一个替代，其中数字信号处理接口部件被拆除到只剩下分组处理，无需更昂贵的话音压缩和回波消除电路。通过将二次复杂度放置到低成本 TDM 交换机中并且将线性复杂度放置在高成本部件中，本发明获得与高端口密度成比例的非常好的经济性。

虽然利用上面的详细描述特别示出了描述了本发明，但是本领域技术人员应当理解，可以在不偏离本发明精神和范围的条件下在形式和细节上作出各种改变、修改、变更、变化和派生。


34
DLC
CB
36
16
PSTN
13
VPS
10
30
MG
类型5
交换机
32
34
DLC
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VPS
ATM
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MG
MG
22
31
40
40
24
DSLAM
ATM
ATM
18
VPS
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42
14
DSLAM
44
IAD
22
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IP
26
42
IAD
20
46

图 1

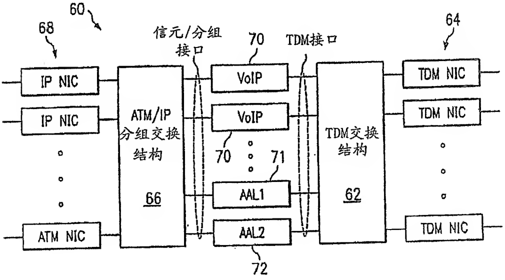
60
68
信元/分组
接口
70
TDM接口
64
IP NIC
IP NIC
ATM NIC
ATM/IP
分组交换
结构
66
VoIP
VoIP
70
71
AAL1
AAL2
72
TDM交换
结构
62
TDM NIC
TDM NIC
TDM NIC

图 2

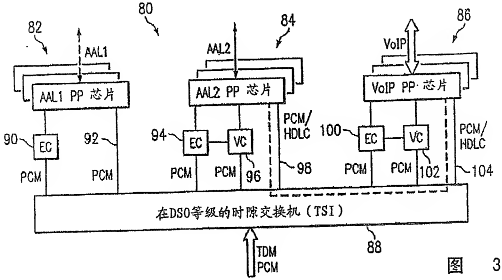
80
82
84
86
AAL1
AAL2
VoIP
AAL1 PP 芯片
AAL2 PP 芯片
VoIP PP 芯片
90
EC
92
94
EC
VC
96
PCM/
HDLC
98
100
EC
VC
102
PCM/
HDLC
104
PCM
PCM
PCM
PCM
PCM
PCM
在DSO等级的时隙交换机（TSI）
88
TDM
PCM

图 3

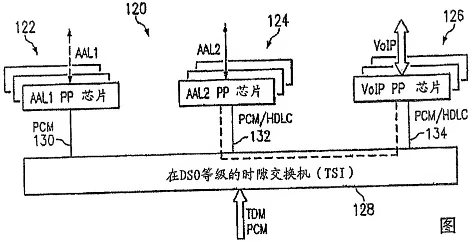
120
122
124
126
AAL1
AAL2
VoIP
AAL1 PP 芯片
AAL2 PP 芯片
VoIP PP 芯片
PCM
130
PCM/HDLC
132
PCM/HDLC
134
在DSO等级的时隙交换机（TSI）
128
TDM
PCM

图 4

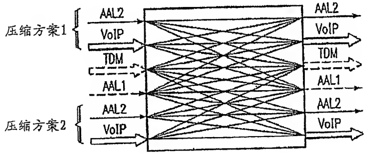
压缩方案1
压缩方案2
AAL2
VoIP
TDM
AAL1
AAL2
VoIP
AAL2
VoIP
TDM
AAL1
AAL2
VoIP

图 5

US007006489B2

(12) **United States Patent**
Li et al.

(10) **Patent No.: US 7,006,489 B2**
(45) **Date of Patent: Feb. 28, 2006**

(54) **VOICE PACKET SWITCHING SYSTEM AND METHOD**

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(73) Assignee: **Santera Systems, Inc.**, Plano, TX (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 850 days.

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(22) Filed: **Feb. 23, 2001**

(65) **Prior Publication Data**

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(51) **Int. Cl.**
H04L 12/66 (2006.01)

(52) **U.S. Cl.** **370/352**; 370/466

(58) **Field of Classification Search** 370/352, 370/353, 400, 401, 465, 466, 469
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,375,121 A 12/1994 Nishino et al.
5,905,873 A * 5/1999 Hartmann et al. 709/249
6,339,594 B1 * 1/2002 Civanlar et al. 370/352
6,731,627 B1 * 5/2004 Gupta et al. 370/352

FOREIGN PATENT DOCUMENTS

DE 198 29 822 A1 1/2000

OTHER PUBLICATIONS

M. Laurence, et al., "Voice Over ATM: A Hybrid TDM/ATM Design Approach," Electronic Engineering, Morgan-Grampian Ltd., vol. 71, No. 869, Jun. 1999, pp. 81-82, 84, 86.
Josef Singer, et al., "Narrowband Services Over ATM Networks: Evaluation of Trunking Methods," World Telecommunications Congress, vol. 2, Sep. 21, 1997, pp. 17-25.

* cited by examiner

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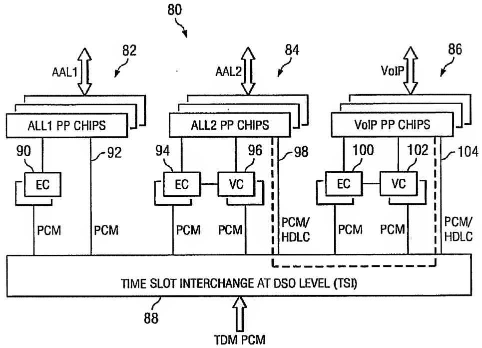
Assistant Examiner—Donald Mills
(74) *Attorney, Agent, or Firm*—Haynes and Boone, LLP

(57) **ABSTRACT**

A switching system includes at least one packet processing circuit interfacing with at least one packet transmission link. The switching system further includes a switch fabric coupled to the at least one packet processing circuit, whereby the switch fabric is operable to switch between channels receiving and transmitting data over the at least one packet transmission link and channels receiving and transmitting data over at least one non-packet transmission link interfaced by the switch fabric.

26 Claims, 3 Drawing Sheets



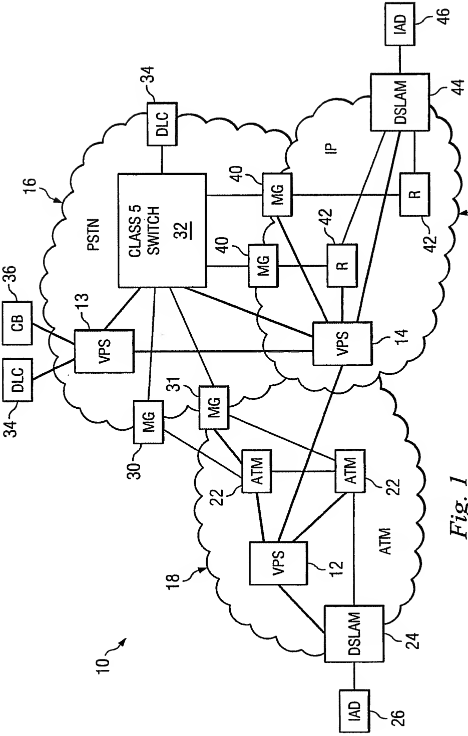

10
16
PSTN
CLASS 5 SWITCH
32
DLC
34
CB
36
VPS
13
34
DLC
MG
30
MG
31
40
MG
40
MG
IP
R
42
R
42
42
20
DSLAM
44
IAD
46
VPS
14
ATM
22
ATM
22
22
18
VPS
12
ATM
DSLAM
24
IAD
26

Fig. 1

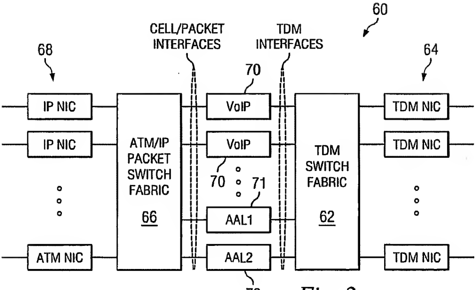

60
CELL/PACKET INTERFACES
TDM INTERFACES
68
64
70
IP NIC
IP NIC
ATM NIC
ATM/IP PACKET SWITCH FABRIC
66
VoIP
VoIP
70
71
AAL1
AAL2
72
TDM SWITCH FABRIC
62
TDM NIC
TDM NIC
TDM NIC

Fig. 2

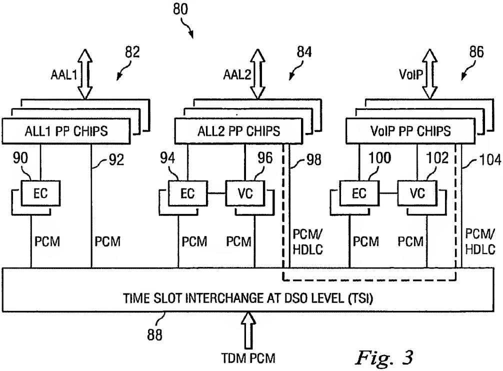

80
AAL1
82
AAL2
84
VoIP
86
ALL1 PP CHIPS
ALL2 PP CHIPS
VoIP PP CHIPS
90
92
94
96
98
100
102
104
EC
EC
VC
EC
VC
PCM
PCM
PCM
PCM
PCM/ HDLC
PCM
PCM
PCM/ HDLC
TIME SLOT INTERCHANGE AT DSO LEVEL (TSI)
88
TDM PCM

Fig. 3

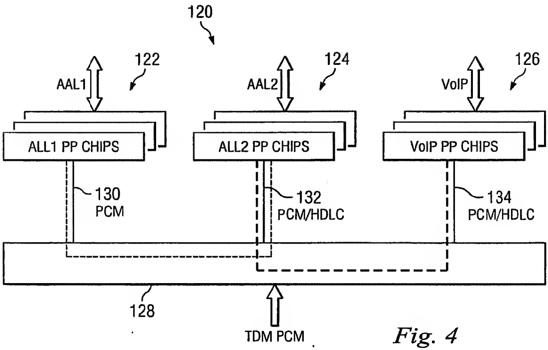

120
122
124
126
AAL1
AAL2
VoIP
ALL1 PP CHIPS
ALL2 PP CHIPS
VoIP PP CHIPS
130
PCM
132
PCM/HDLC
134
PCM/HDLC
128
TDM PCM

Fig. 4

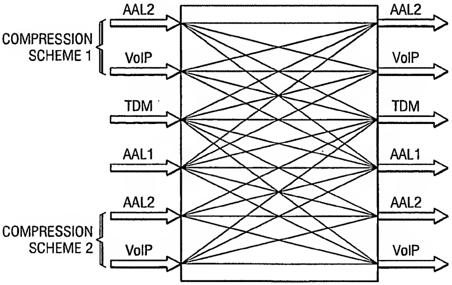

AAL2
COMPRESSION SCHEME 1
VoIP
TDM
AAL1
AAL2
COMPRESSION SCHEME 2
VoIP
AAL2
VoIP
TDM
AAL1
AAL2
VoIP

Fig. 5

VOICE PACKET SWITCHING SYSTEM AND METHOD

TECHNICAL FIELD OF THE INVENTION

This invention relates to telecommunication equipment, and more particularly, to voice packet switching system and method.

BACKGROUND OF THE INVENTION

The current trend in telecommunications network evolution focuses on supplanting the traditional public switched telephone network (PSTN) with emerging packet-switched based data networks. There are a number of contributing factors to this trend. Data traffic volume has far exceeded voice traffic volume and the gap continues to grow unabated. Studies have shown that voice will constitute only a very small percentage of the total telecommunication traffic. Furthermore, a data network is inherently more flexible for supporting a variety of voice and data services and has the technical potential to support all the services that are carried over PSTN today.

It will take years before all technical issues are resolved so that data networks can have the superior quality and reliability of PSTN. Meanwhile, PSTN and the rapidly developing data networks will continue to coexist. To facilitate the migration of voice telephony from PSTN toward packet networks, and to make two networks interoperate for voice services, network operators are deploying voice media gateways. Voice media gateways convert voice signals between time division multiplex (TDM) pulse code modulation (PCM) and a data packet format, either asynchronous transfer mode (ATM) or voice over Internet protocol (VoIP). Therefore, voice media gateways are currently used to bridge the data network and the PSTN, but does not perform the necessary switching functions. This type of transition network, though necessary, is not economical since it introduces duplicative network elements and increases the cost of operating and maintaining the network.

SUMMARY OF THE INVENTION

In accordance with the present invention, a voice packet switch provides end-to-end voice switching regardless of the encoding, multiplexing, or transmission protocols used by the calling and called parties, and the inter-switch trunks along the voice connection's path. The voice packet switch is further operable with all voice encoding and compression schemes, physical layer transmission protocols, access protocols, and trunking protocols. An important aspect of the present invention is the decoupling of the three key functions in switching voice over packet-switched networks: packet processing, digital signal processing, and switching. The voice packet switch of the present invention allows each key function to be implemented in the most cost efficient manner for high density applications, such as in the central office. Furthermore, each function may be implemented with the most advanced state of the art technology independently from one another. Another benefit from eliminating echo canceling and voice compression/decompression functionality from the voice packet switch is improved voice fidelity and echo cancellation quality. Further, significant end-to-end delays in voice transmission associated with jitter buffering are also eliminated. The key is the elimination of the voice compression/decompression cycles on the intermediate switches along the path of a voice connection.

In one embodiment of the present invention, a switching system includes at least one packet processing circuit interfacing with at least one packet transmission link. The

switching system further includes a switch fabric coupled to the at least one packet processing circuit, whereby the switch fabric is operable to switch between channels receiving and transmitting data over the at least one packet transmission link and channels receiving and transmitting data over at least one non-packet transmission link interfaced by the switch fabric.

In yet another embodiment of the present invention, a switching system includes at least one packet processing circuit interfacing with first data traffic in a first protocol received and transmitted using a first transmission protocol. The switching system further includes a switch fabric coupled to the at least one packet processing circuit, whereby the switch fabric is operable to switch between channels coupled to the first data traffic and channels coupled to a second data traffic in a second protocol received and transmitted using a second transmission protocol interfaced by the switch fabric.

In yet another embodiment of the present invention, a voice packet switch includes a plurality of ATM packet processing circuits operable to receive and transmit data packets over a plurality of ATM transmission links and operable to extract data from the ATM data packets and insert the extracted data into data frames of a predetermined protocol. The voice packet switch further includes a time slot interchange coupled to the plurality of ATM packet processing circuits and operable to switch the data frames received therefrom and multiplexed PCM data received and transmitted over a plurality of PCM transmission links.

In yet another embodiment of the present invention, a voice packet switch includes a plurality of VoIP packet processing circuits operable to receive and transmit data packets over a plurality of IP transmission links and operable to extract data from the IP data packets and insert the extracted data into data frames of a predetermined protocol. The voice packet switch further includes a time slot interchange coupled to the plurality of VoIP packet processing circuits and operable to switch the data frames received therefrom and multiplexed PCM data received and transmitted over a plurality of PCM transmission links.

In yet another one embodiment of the present invention, a method of switching voice packets includes the steps of receiving data packets transmitted over at least one packet transmission link, processing the data packets, and converting the data packets to data frames of a predetermined data link protocol. The data frames are received at an input channel of a switch fabric, and output to an output channel of the switch fabric. The data frames are then converted to data packets, and transmitted over at least one packet transmission link.

BRIEF DESCRIPTION OF THE DRAWINGS

For a more complete understanding of the present invention, the objects and advantages thereof, reference is now made to the following descriptions taken in connection with the accompanying drawings in which:

FIG. 1 is a simplified block diagram of an evolving telecommunication network employing voice packet switching systems of the present invention;

FIG. 2 is a simplified architectural block diagram of an embodiment of the voice packet switching system according to the teachings of the present invention;

FIG. 3 is a simplified block diagram of an alternative embodiment of a voice packet switching system according to the teachings of the present invention;

FIG. 4 is a simplified block diagram of another embodiment of a voice packet switching system according to the teachings of the present invention; and

FIG. 5 is a graphical representation of the any-to-any switching functionality of the voice packet switching system and a method for grouping traffic with the same voice compression encoding scheme.

DETAILED DESCRIPTION OF THE DRAWINGS

The preferred emhodiment of the present invention and its advantages are hest understood hy referring to FIGS. 1 through 5 of the drawings, like numerals heing used for like and corresponding parts of the various drawings.

FIG. 1 is a simplified hlock diagram of an evolving telecommunication network 10 employing voice packet switching systems 12–14 of the present invention. Evolving network 10 is a conglomerate of current and next generation equipment operating in a puhlic switched telephone network (PSTN) 16 for voice services, and an asynchronous transfer mode network (ATM) 18 and an Internet protocol network (IP) 20 for data services. ATM is a high bandwidth, low-delay, connection-oriented, packet-like switching and multiplexing technology used to support the transmission of data, voice data, video data, image data, multimedia data traffic. ATM network 18 employs equipment such as ATM switches 22, digital subscriber line access multiplexers (DSLAM) 24, and integrated access devices (IAD) 26. ATM network 18 further includes one or more voice packet switches 12 of the present invention for switching individual multiplexed voice streams within the ATM network. A voice stream is defined herein as unidirectional successive voice signals from one party to another in the same voice connection.

ATM network 18 is coupled to PSTN 16 via voice media gateways (MG) 30 and 31. PSTN employs equipment such as Class 5 switches 32, Class 4 switches (not shown), digital loop carriers 34, and channel banks (CB) 36 to route and deliver voice and data traffic. PSTN 16 is further coupled to IP network 20 via voice media gateways 40 to convert the voice signals between TDM PCM and IP. IP network 20 additionally employs equipment such as routers 42, DSLAMs 44, and IADs 46 in addition to a voice packet switching system 14 of the present invention. Using the voice packet switching systems of the present invention, voice traffic can be carried over any data packet protocol (VoX), such as IP and ATM protocols favored today.

It may be seen that a voice packet switch may function as an interface device between two networks of disparate formats, such as voice packet switch 14 between ATM and IP networks 18 and 20. The existence of voice media gateways in evolving network 10 is due to their integration into the telecommunications network to solve the voice-data interface problem prior to the introduction of the voice packet switching system. With the deployment of voice packet switching systems, the voice media gateways may he eventually phased out as network deployment logistics and capital equipment schedules allow.

The voice packet switching system of the present invention is operahle to perform any-to-any switching so that a voice stream can be switched between any encoding and transmission schemes. The voice packet switch is operahle to terminate the inhound channel, extract the voice signal carried within and put it hack on the matching outhound channel, regardless of whether the same or different encoding and transmission schemes are used hy the inbound and outhound channels. Furthermore, these functionalities are performed economically even at a large scale. Because there is a fundamental difference hetween TDM (time division multiplex) to AAL1 (ATM adaptation layer type 1) and AAL2 (ATM adaptation layer type 2) to VoIP (voice over IP) in the ways individual voice streams are multiplexed over a single connection, the voice packet switch of the present invention differentiates and operates differently on them. Each TDM channel or AAL1 connection carries only one voice stream, while each AAL2 connection may carry up to 240 simultaneous voice streams, and each VoIP connection may carry even more. The voice packet switching system of the present invention is operable to de-multiplex the voice streams and switch them individually to their corresponding outbound channels according to the path of each voice conversation, and then re-multiplex all the outbound voice streams sharing the same next hop switch onto the same outhound AAL2 or VoIP connections. In this way, voice applications can he decoupled from the specifics of the encoding and transmission schemes that are of little concern or consequence to the end users. Finally, as a central office switch for network carriers, the voice packet switch of the present invention is operable to he scaled up economically as the density of voice ports increases.

The current state of the art does not include any equipment that satisfies the above-enumerated requirements. For example, media gateways only perform voice conversions either between TDM and ATM or hetween TDM and VoIP, but seldom both. Further, media gateways do not perform switching functions. No IP router or ATM switch today perform the individual voice stream switching when these streams are multiplexed. All ATM switches today perform switching in ATM format only, and IP routers switch IP packets between different IP ports only. No multiplexing of voice streams is performed. Further, IADs and DLCs convert voice streams between IP or ATM and TDM only, and typically do not perform any switching functionality. It will he seen with reference to FIGS. 2–5 below that the voice packet switching system and method of the present invention provides a unique solution to converge voice and data communications into tomorrow's data-centric telecommunications network.

FIG. 2 is a simplified base architectural block diagram of an embodiment of the voice packet switching system 60 according to the teachings of the present invention. Voice packet switching system 60 includes a TDM switch fabric 62, which is coupled to TDM transmission links via TDM network interface controllers (NICs) 64. TDM switch fabric 62 is a time slot interchange with a channel size of DS0 or 64 Kbps bandwidth. An ATM/IP switch fabric 66 is operable to switch variable length packets. Network interface controllers 68 coupled to ATM/IP switch fabric 66 is operable to terminate IP or ATM transmission links. Voice service modules such as VoIP 70, AAL1 71 and AAL2 72 are circuits or chips that are operable to perform interfacing, packet processing, and optionally voice compression and echo cancellation. AAL1 supports Class A traffic, which is connection-oriented, constant bit rate (CBR), time-dependent traffic such as uncompressed digitized voice and video data. AAL2 supports Class B traffic, which is connection-oriented, variable bit rate (VBR), isochronous traffic requiring precise timing between source and sink such as compressed voice and video traffic. Although these voice service modules are shown as separate logical blocks in FIG. 2, they may be integrated and realized on a single circuit board or in a variety of configurations. Switch 60 is operahle to perform any-to-any switching between voice and data formats and speeds.

FIG. 3 is a simplified block diagram of an emhodiment of a voice packet switching system 80 according to the teachings of the present invention. Voice packet switching system 80 is a simple and straightforward implementation of the base architecture shown in FIG. 2. Voice packet switching system 80 includes AAL1 packet processing (PP) circuits or chips 82, AAL2 packet processing circuits or chips 84, and VoIP packet processing circuits or chips 86 interfacing and

processing the AAL1, AAL2, and VoIP packets, respectively. AAL1 packet processing circuits 82 are coupled to a time slot interchange 88 or the TDM switch fabric via echo canceling circuits or chips 90 and a pulse code modulation link 92. AAL2 packet processing circuits 84 are coupled to a time slot interchange 88 via echo canceling circuits 94, voice compression circuits or chips 96, and a pulse code modulation/high-level data link control (PCM/HDLC) link 98. HDLC is a link layer protocol standard for point-to-point and point-to-multipoint communications. HDLC encapsulates packet data in a frame with the frame header and trailer including various control information such as error control mechanisms. HDLC variants include frame relay, link access procedure-balanced (LAP-B), link access procedure-data channel (LAP-DC), point-to-point protocol (PPP), and synchronous data link control (SDLC).

Using the data link layer protocol, one bit-stream HDLC channel per voice connection is used to interconnect two AAL2 and VoIP packet processing circuits through time slot interchange 88. The encoded voice frames (or mini packets) encapsulated in AAL2 cells or VoIP packets are extracted and put in the HDLC frames for switching through time slot interchange 88. After switching, these voice frames are put back to the outbound AAL2 cells or VoIP packets. Silence insertion descriptor (SID) messages for silence detection purpose can be transparently passed through AAL2 packet processing circuit 84 between a standard ATM interface and time slot interchange 88, using different user-to-user indication (UUI), as to propagate through the network. AAL2 packet processing circuit 84 does not perform the service specific convergence sublayer (SSCS) functions, so both SID messages and voice packets are treated identically. HDLC channel overhead includes two bytes for AAL2 header, two bytes for the HDLC header, and bit stuffing, while the total HDLC channel raw bandwidth is limited to 64 kbps. If PCM is used to carry a voice stream over AAL1, the PCM stream can run through in their native DS0 TDM format instead of using an HDLC channel. HDLC protocol is selected in this embodiment because it is widely implemented and many commercial off-the-shelf hardware and software are readily available. Other suitable protocols may be used if additional benefits can be conferred. After compression, a voice stream is no more than 40 Kbps. The total sum of the voice payload, the HDLC overhead, and AAL2/VoIP overhead is under 64 Kbps. This crucial observation urges the use of time slot interchange 88 as the unified switch fabric to meet all voice stream switching requirements.

In operation, switching between a pair of TDM traffic is preferably done in PCM. Switching between a pair of ATM AAL1 is preferably done in PCM. Any combination of AAL2/VoIP to AAL2/VoIP switching, both with the same adaptive differential pulse code modulation (ADPCM) encoding subtype, can be switched using direct PCM or HDLC. If direct PCM is chosen, only a fixed subset of bits within each byte of the encoded voice sample is filled, processed, and utilized for playback. This option requires the activation of jitter buffering. If HDLC is chosen, no jitter buffering is required. Jitter buffering is used to smooth out the random packet delay variations as the packets travels hop by hop through the telecommunications network. Because of the random variations of the time interval between the arrival of two successive cells or packets from the same voice conversation, each switch is required to assume the worst-case delay and provision large jitter buffers. As a result, the cumulative transmission delay is very significant for a voice connection that spans many hops, as is typical for toll calls. Further, this large delay makes satisfactory echo cancellation difficult, and seriously impairs the subjective quality, or the naturalness, of a voice conversation. Therefore, the HDLC path is preferred over PCM. Any combination of AAL2/VoIP to AAL2/VoIP with the same, non-ADPCM, compression encoding is preferably switched using HDLC. Any combination of AAL2/VoIP to AAL2/VoIP with different compression coding is preferably done by first converting the traffic to PCM before switching. For AAL1/TDM to AAL2/VoIP with voice compression, it is also preferable to first convert to PCM before switching. If voice compression is not used at all, switching is preferably done in PCM. The goal is to avoid repeated conversions to PCM at intermediate hops if possible by utilizing any one of the standardized data link protocols to encapsulate and transport compressed voice streams on the uniform PCM channels through time slot interchange 88. The preferred properties of the chosen data link protocol include acceptance of variable length voice mini packets, a total bandwidth not to exceed 64 Kbps, and error detection and recovery capabilities, such as HDLC and similar protocols.

If the option of using the standardized data link protocol to transport compressed voice streams on the uniform PCM channels is not available, conversion to PCM must be performed at each hop, requiring jitter buffering at each hop. Further, each compression and decompression cycle introduces certain amount of distortion. The accumulation of such distortion as a voice connection traverses multiple switches can be substantial for additional impairment of the voice qualities. A further disadvantage is the prohibitively high cost or limitations on heat dissipation associated with integrated packet processing, echo canceling and voice compression chips or modules. An integrated chip is also ill suited for a central office application because of high unit cost, low density, and heat dissipation concerns.

The present invention becomes even more relevant in view of the current trend toward the predominance of AAL2/VoIP for voice traffic in next generation networks. In these networks, the process of performing repetitive echo cancellation and voice compression/decompression at each network hop becomes glaringly inefficient, unnecessary and disadvantageous (due to quality degradation).

FIG. 4 is a simplified block diagram of another embodiment of a voice packet switching system 120 that has been decoupled from digital signal processing according to the teachings of the present invention. Voice packet switching system 120 includes no costly digital signal processing circuits or chips and therefore does not perform echo cancellation or voice compression. A voice connection path consists of one originating switch, one terminating switch, and intermediate switches. When voice compression is used, it only has to take place in the originating switch and the terminating switch, and is unnecessary in the intermediate switches. Further motivation to move digital signal processing out of the voice packet switch is the decreasing economic and technological incentives to compress voice because of the dramatic and continued increase in network bandwidth, the explosive growth of data traffic, and the diminishing percentage of voice in the overall traffic mix. Many mandatory digital signal processing functions as well as the optional voice compression function may be moved to the customer premises equipment (CPE), because integrated chip sets that incorporate packet processing, echo cancellation and voice compression functions are readily available. These chip sets, while ill suited for central office switches, are ideal in CPE applications where highly integrated functionality is valued but density is seldom a serious consideration.

As shown in FIG. 4, voice packet switching system 120 includes AAL1 packet processing circuits 122, AAL2 packet processing circuits 124 and VoIP packet processing circuits 126. AAL1 packet processing circuits 122 are coupled to a

time slot interchange 128 via a PCM data link 130. AAL2 packet processing circuits 124 are coupled to time slot interchange 128 via a PCM/HDLC data link 132. VoIP packet processing chips 126 are coupled to time slot interchange 128 via a PCM/HDLC data link 134. Traffic is presented to and switched by time slot interchange 128 either in native PCM format for uncompressed voice streams or in a standard data link protocol such as HDLC for compressed voice streams. The TDM time slot interchange 128 is an integral component of the overall architecture of voice packet switching system 120 to facilitate the implementation of the requisite Class 5 functions such as conference call bridging, Communications Assistance for Law Enforcement Act. (CALEA), operator barge-in, tone generation, digital announcements, and in-band voice signaling terminations and dialed digits.

By provisioning and/or configuring the network, a network operator may set up separate groups of AAL2 or VoIP connections on both the access links and the trunks, with each group provisioned for voice channels of a single profile. A profile is a scheme governing how voice can be carried over IP packets and ATM AAL2 cells. The profile defines what compression encoding is to be used, the format of the voice samples or frames, and how these samples or frames are to be encapsulated into the IP/RTP (Internet protocol/realtime transport protocol) packets or ATM AAL2 cells. A voice connection set-up request will carry a parameter specifying the choice of the profile. In setting up a voice connection path, the control software on each voice packet switch along the way will pick an appropriate matching outbound channel from the group with the same profile as that from the inbound channel. As a result, the entire path of the voice connection will use the same profile, or the same compression encoding. This provisioning concept is shown in FIG. 5.

Note that the proposed partitioning by profile does not require network topological changes, nor does it necessarily affect the routing of voice calls. The segregation or partitioning is on the logical connection level only. On a physical link joining two adjacent voice packet switches, for example, two or more AAL2/VoIP connections for different encoding can be provisioned prior to the acceptance of voice calls.

Because the DSP-less voice packet switch architecture leads to reduced footprint size and/or increased port density, costs associated with housing and heat dissipation for the switch is greatly reduced. The net benefits to the network operator are much lower capital outlay and ongoing operating costs in terms of real estate, power consumption, and air conditioning expenses. The greatly simplified hardware architecture also dramatically increases the intrinsic reliabilities of the overall switch system. Because the costs associated with DSP is significantly higher than the costs associated with packet processing and switching, the voice packet switch of the present invention achieves the best economy of scale at any given density.

The voice packet switch of the present invention achieves end-to-end voice switching regardless of the encoding, multiplexing, or transmission schemes used by the calling and called parties, and the inter-switch trunks along the voice connection's path. The voice packet switch is further operable with all voice encoding and compression schemes, physical layer transmission protocols, access protocols, and trunking protocols.

An important aspect of the present invention is the decoupling of the three key functions in switching voice over packet-switching networks: packet processing, digital signal processing (echo cancellation and voice compression), and switching. In doing so, the voice packet switch of the present invention allows each key function to be implemented in the most cost efficient manner for high density applications, such as in the central office. Furthermore, each function may be implemented with the most advanced state of the art technology independently from one another. Another benefit from eliminating DSP functionality from the voice packet switch is improved voice fidelity and echo cancellation quality. Further, significant end-to-end delays in voice transmission associated with jitter buffering are also eliminated.

The cost benefits provided by the present invention can be quantified or estimated. For a switch with N ports, there are N-port equivalent of interface circuits and N^2—port equivalent of switching cross points. The cost per port, which is of the most concern to a service provider, is the total cost divided by N. Therefore, the per port cost is the sum of the per port interface cost and the cost of N cross points. In the preferred embodiment of the present invention, the TDM switch fabric can be realized at very low costs because it is a scaling of simple and mature technology. The present invention first reduces the need for voice compression/decompression and echo canceling, and then offers another alternative where the digital signal processing interface components have been stripped down to only packet processing without the much more expensive voice compression and echo canceling circuits. By placing the quadratic complexity in the low cost TDM switch and the linear complexity in high cost components, the present invention achieves very good economics in scale to high port density.

While the invention has been particularly shown and described by the foregoing detailed description, it will be understood by those skilled in the art that various changes, alterations, modifications, mutations and derivations in form and detail may be made without departing from the spirit and scope of the invention.

What is claimed is:

1. A voice packet switch, comprising:

a plurality of ATM packet processing circuits operable to receive and transmit data packets over a plurality of ATM transmission links and operable to extract data from the ATM data packs and insert the extracted data into data frames of a predetermined protocol; and

a time slot interchange coupled to the plurality of ATM packet processing circuits and operable for DS0-level and other per-voice-channel-level switching of;

data frames received from the ATM packet processing circuits; and

multiplexed PCM data received and transmitted over a plurality of PCM transmission links; wherein

the voice packet switch is configured for operation as;

a PSTN network switching element; and

a network gateway bridging a PSTN network and a packet-based network; and

the voice packet switch is deployed as;

the PSTN network switching element; or

the network gateway.

2. The voice packet switch, as set forth in claim 1, wherein the plurality of ATM packet processing circuits comprise AAL1 packet processing circuits operable to receive and transmit AAL1 cells.

3. The voice packet switch, as set forth in claim 1, wherein the plurality of ATM packs processing circuits comprise AAL2 packet processing circuits operable to receive and transmit AAL2 cells and operable to extract data from the AAL2 cells and inert the extracted data into HDLC data frames for switching by the time slot interchange.

4. The voice packet switch, as set forth in claim 1, wherein the plurality of ATM packet processing circuits comprise VoIP packet processing circuit, operable to receive and

transmit VoIP data packets and operable to extract data from the VoIP data packets and insert the extracted data into HDLC data frames for switching by the time slot interchange.

5. The voice packet switch, as set forth in claim 1, further comprising at least one digital signal processing circuit coupled between the plurality of packet processing circuits and the time slot interchange operable to convert between ATM data packets and multiplexed PCM data.

6. The voice packet switch, as set forth in claim 5, wherein the at least one digital signal processing circuit comprises at least one echo canceling circuit.

7. The switching system, as set forth in claim 5, wherein the at least one digital signal processing circuit comprises at least one voice compression circuit.

8. A switching system, comprising:

at least one packet processing circuit configured to process non-TDM data packets received and transmitted over at least one packet on link; and

a switch fabric having:

at least one first channel configured to receive and transmit the non-TDM data packets over the at least one packet transmission link; and

at least one second channel configured to receive and transmit multiplexed TDM data over at least one non-packet transmission link;

wherein the switch fabric is operable for DS0-level another per-voice channel level switching between the at least one first channel and the at least one second channel, wherein the switching stem is configured for operation as: PSTN network switching element; and a network gateway bridging a PSTN network and a packet-based network; and wherein the switching system is deployed as: the PSTN network switching element; or the network gateway.

9. The switching system, as set forth in claim 8, wherein the at least one packet processing circuit is operable to extract the non-TDM data packets received over the at least one packet transmission link and transmit to the switch fabric using a data link protocol.

10. The switching system, as let forth in claim 8, wherein the at least one packet processing circuit is operable to extract compressed voice data received over the at least one packet transmission link and transmit to the switch fabric for switching using a data link protocol.

11. The switching system, as set forth in claim 8, wherein the at least one packet processing circuit is operable to extract AAL2 voice data received over the at least one packet transmission link and transmit to the switch fabric for switching using high-level data link control protocol.

12. The switching system, as set forth in claim 8, wherein the at least one packet processing circuit is operable to extract VoIP voice data received over the at least one packet transmission link and transmit to the switch fabric for switching using high-level data link control protocol.

13. The switching system, as set forth in claim 8, wherein the at least one packet processing circuit is operable to extract data received from the switch fabric and insert the extracted data into packets for transmission over the at least one packet transmission link.

14. The switching system, as set forth in claim 8, wherein the at least one packet processing circuit is operable to extract voice data received from the switch fabric and insert the extracted voice data into AAL2 cells for transmission over the at least one packet transmission link.

15. The switching system, as set forth in claim 8, wherein the at least one packet processing circuit is operable to extract voice data received from the switch fabric and insert the extracted voice data into VoIP packets for transmission over the at least one packet transmission link.

16. The switching system, as set forth in claim 8, wherein the switch fabric is a time slot interchange switch fabric.

17. The switching system, as set forth in claim 8, wherein the switch fabric is operable to switch multiplexed uncompressed voice data.

18. The switching system, as set forth in claim 8, wherein the switch fabric is operable to switch PCM data.

19. The switching system, as set forth in claim 8, wherein the at least one packet transmission link includes a plurality of packet transmission links, including a multiplexed VoIP data transmission link and a multiplexed AAL2 data transmission link.

20. The switching system, as set forth in claim 8, wherein the at least one packet processing circuit includes at least one AAL1 packet processing circuit operable to interface with at least one ATM transmission link.

21. The switching system, as set forth in claim 8, wherein the at least one packet processing circuit includes at least one AAL2 packet processing circuit operable to interface with at least one ATM transmission link.

22. The switching system, as set forth in claim 8, wherein the at least one packet processing circuit includes at least one VoIP packet processing circuit operable to interface with at least one IP transmission link.

23. The switching system, as set forth in claim 8, further comprising at least one digital signal circuit coupled between the at least one packet processing circuit and the switch fabric and operable to convert between a first data form corresponding to the at least one packet transmission link and a first data form corresponding to the at least one non-packet transmission link.

24. The switching system, as set forth in claim 8, further comprising at least one digital signal processing circuit coupled between the at least one packet processing circuit and the switch fabric and operable to convert between a compressed data from corresponding to the at least one packet transmission link and an uncompressed data form corresponding to the at least one non-packet transmission link.

25. The switching system, as set forth in claim 24, wherein the at least one digital signal processing circuit comprises at least one echo canceling circuit.

26. The switching system, as set forth in claim 24, wherein the at least one digital signal processing circuit comprises at least one voice compression circuit.

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UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO. : 7,006,489 B2
DATED : February 28, 2006
INVENTOR(S) : San-Qi Li et al.

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

Column 4,
Line 46, "71" and "72" should be bold.

Column 8,
Line 39, "packs" should read -- packets --.
Lines 43, 48 and 52, the semicolon should be replaced with a colon.
Line 60, "packs" should read -- packet --.
Line 63, "inert" should read -- insert --.
Line 67, "circuit" should read -- circuits --.

Column 9,
Line 19, "on" should read -- transmission --.
Line 28, "another" should read -- and other --.
Line 30, "stem" should read -- system --.
Line 41, "let" should read -- set --.

Column 10,
Line 38, -- processing -- should be between "signal" and "circuit".

Signed and Sealed this

Ninth Day of May, 2006

Jon W. Dudas

JON W. DUDAS
Director of the United States Patent and Trademark Office